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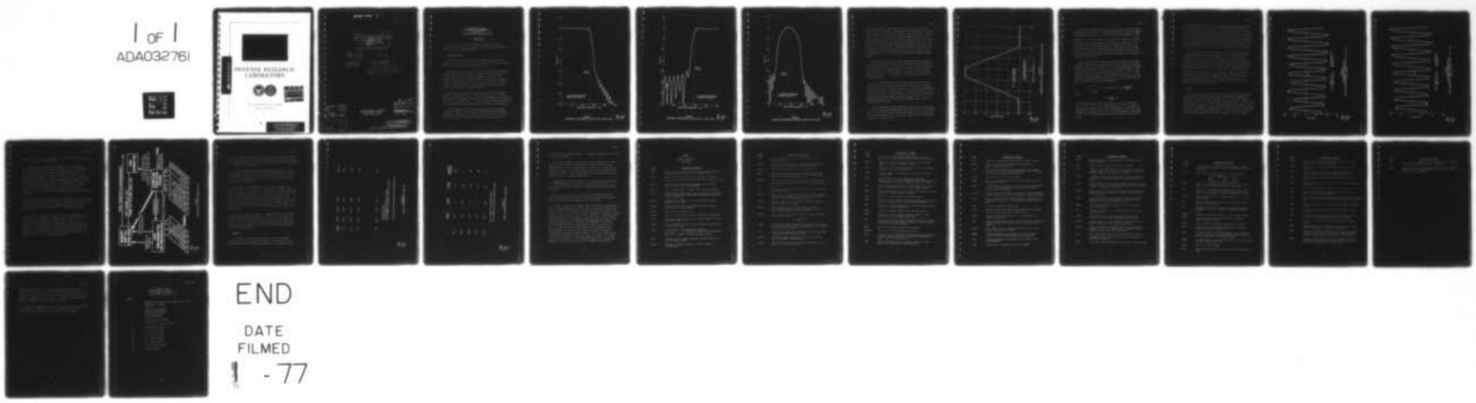

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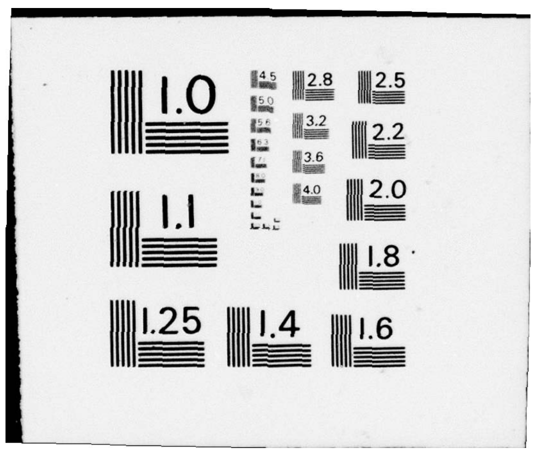
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CONTRACT NObsr-95181
QUARTERLY PROGRESS REPORT NO. 2
for the period
1 November 1966 - 31 January 1967

I. INTRODUCTION

During the second quarter of effort on Contract NObsr-95181, significant steps were taken to complete the background work required for both the Cepstrum and PERT studies.

A. Cepstrum Alanysis

The Cepstrum Alanysis effort on this contract has been diverted during the last quarter into several related areas. This work is preliminary to the Cepstrum Alanysis of some sonar echoes.

The first of these, digital filtering, was discussed in the previous Quarterly Progress Report, 1 August - 31 October 1966. The effort has continued and has been considerably expanded in the areas of more sophisticated design techniques, construction of comb filters, digital filtering of analog data, etc. These studies were called for in order to give us ample capability in the processing of digital data and to provide a second approach to Cepstral Alanysis.

Specifically and briefly, a Cepstrum Alanysis involves a spectrum analysis, a digital filter and a second spectrum analysis. A spectrum analysis can be performed by computing the Fourier transform of a covariance or by the use of a comb filter. It was decided that these techniques should be implemented since these methods are not, in practice, equivalent.

The initial attempts at building digital filters were reasonably successful. Figures 1, 2, and 3 show the frequency response of some of these filters. Problems arose, however, if the filters were either too narrow or too broad. In the case of a narrow filter some trouble is

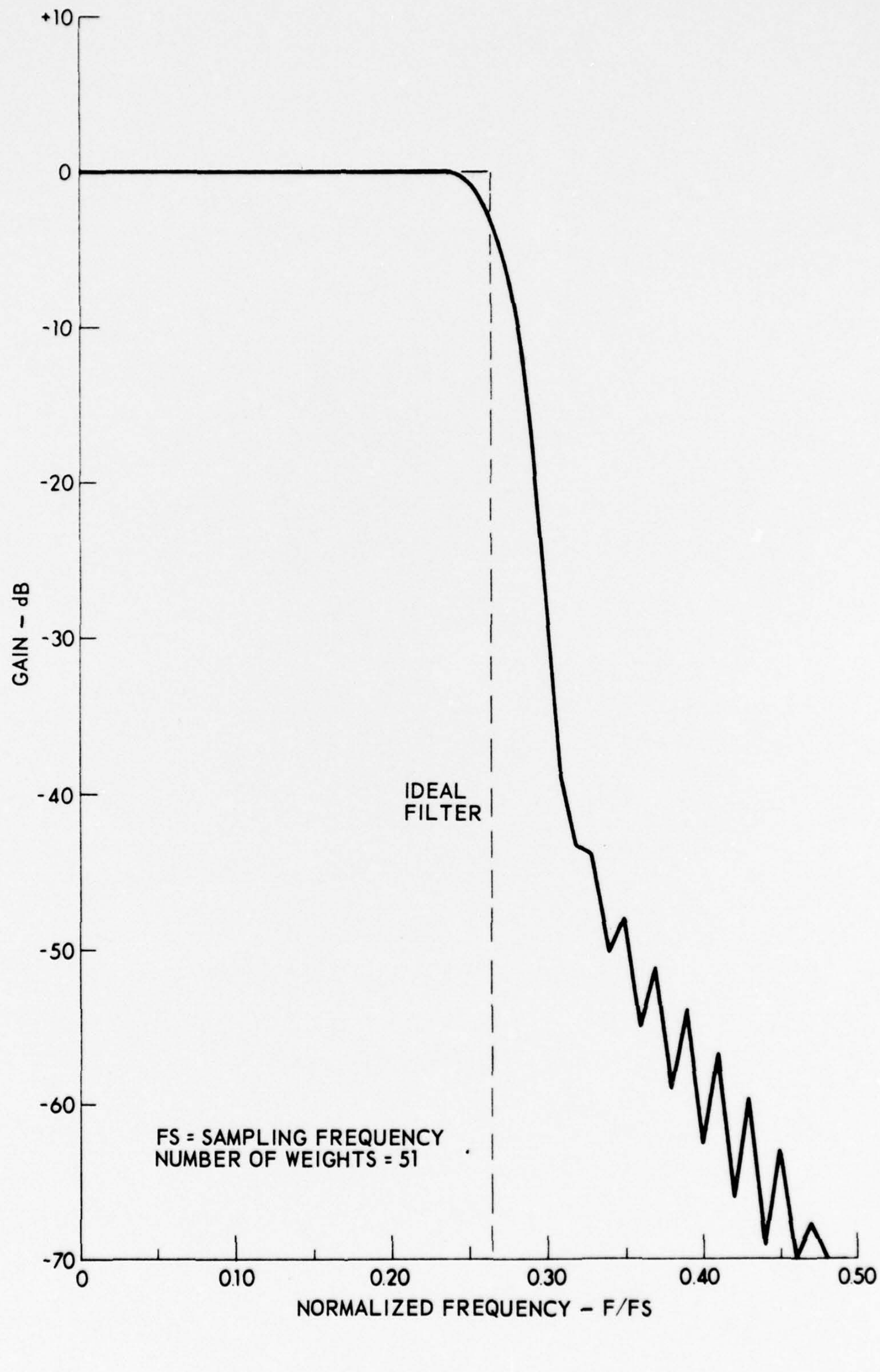

FIGURE 1
FREQUENCY RESPONSE OF A DIGITAL LOW - PASS FILTER

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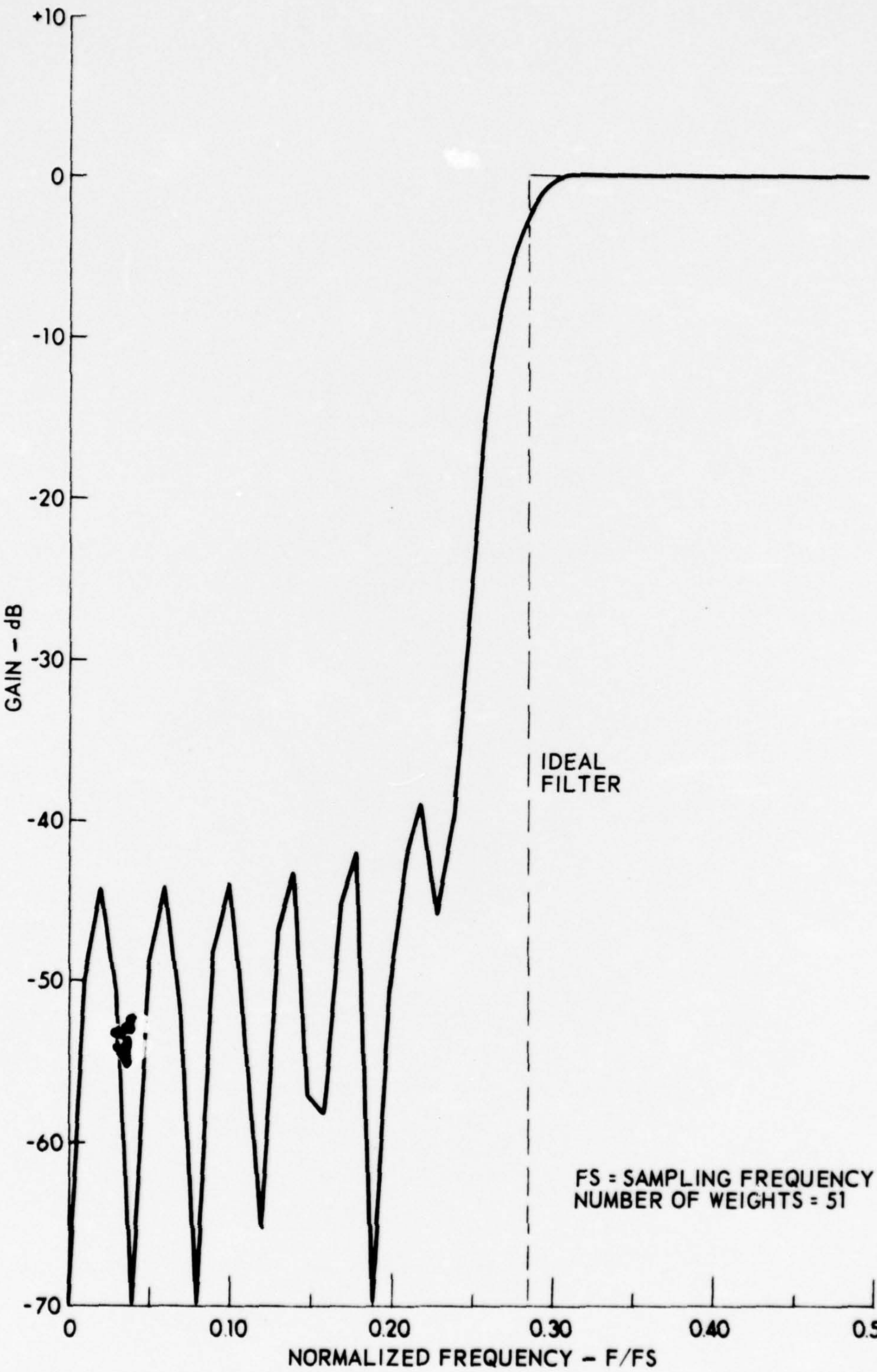

FIGURE 2
FREQUENCY RESPONSE OF A DIGITAL HIGH - PASS FILTER

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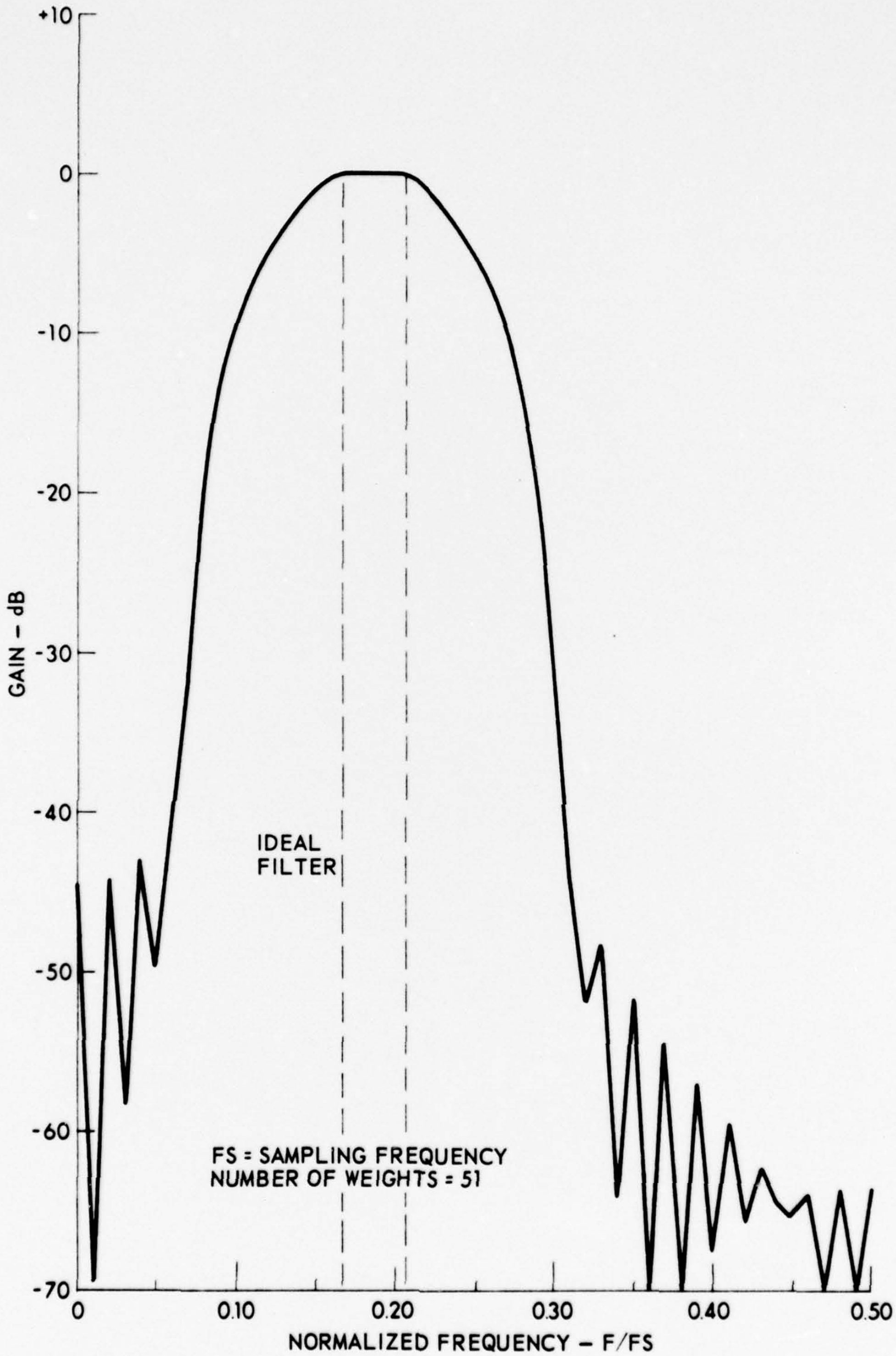

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FIGURE 3
FREQUENCY RESPONSE OF A DIGITAL BAND - PASS FILTER

expected because with a narrow analog filter the rise time is roughly proportional to the inverse of the bandwidth. The problems we encountered, however, seemed to be related to the rate of decay of the filter--the slope of the "skirt". The broad filters (several decades, for example) were a problem because the design criterion uses a linear frequency axis, whereas the filters are normally described on a logarithmic frequency axis. This results in an uneven sampling of the filter with sparse sampling at the low-frequency end as compared to dense sampling at the high-frequency end. The standard solution to this problem for the classical digital filter is to replace the filter with several filters covering the desired frequency range.

It was decided that the more elegant approach to digital filtering--the fast Fourier transform (FFT)--should be implemented and compared with the classical approach. To implement these filters the input wave form is first converted to the frequency domain via the FFT. The data in the frequency response are then filtered by multiplication by the response curve of the filter. The inverse FFT then yields an approximation to the filtered input data.

The FFT approach bypasses several of the problems encountered with classical filters. The design of complicated filters is much simpler in that only the frequency response is required. In general, the FFT approach is much faster because a multiplication in the frequency domain is used in place of a convolution in the time domain. It is not known at present how well an FFT filter approximates a real-world analog filter. A topic of immediate importance is the comparison of the FFT technique with the classical digital filters.

Subroutines making classical digital filters available to users of the CDC 3200 at Defense Research Laboratory (DRL) have been written and checked out. Within the limitations discussed above these filters are quite good. They have been compared with an "equivalent" analog filter. The results are excellent as indicated in Fig. 4

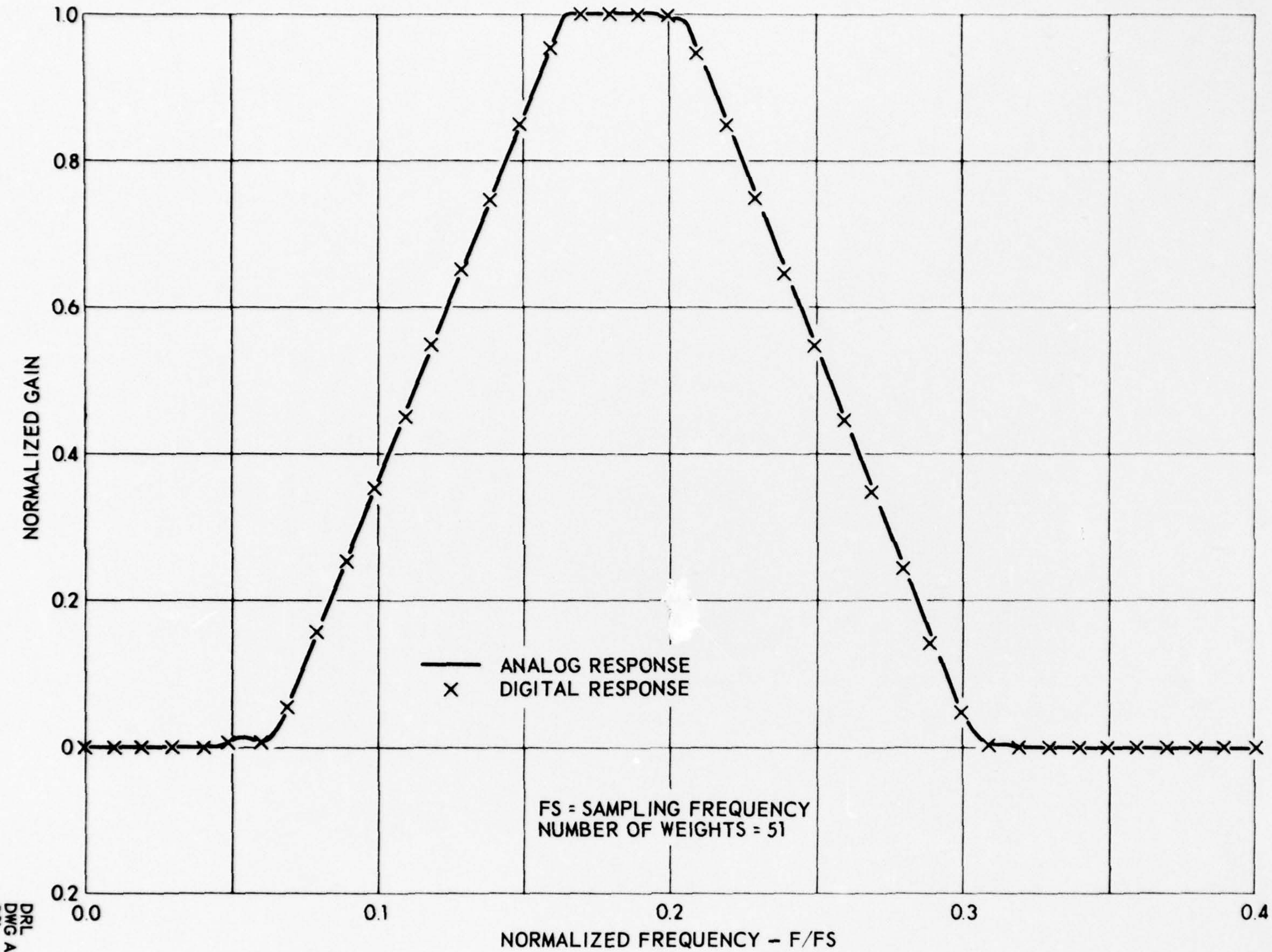

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FIGURE 4
RESPONSE CURVES OF SIMILAR ANALOG AND DIGITAL FILTERS

This brings us rather naturally to another item which has received a great deal of our attention. To introduce the problem, recall that Shannon's Theorem states that if $f(t)$ is band-limited with center frequency ω_o and bandwidth B; i.e., the spectrum of f lies between $\omega_o - \frac{B}{2}$ and $\omega_o + \frac{B}{2}$, then $f(t)$ can be recovered from samples of f taken at the frequency 2B. The conditions of this theorem are never met in practice and, therfore, we find that we are trying to approximate $f(t)$ from values of f at discrete times. Even more often, we are concerned with further processing of the digitized data such as digital filtering, spectrum analysis, correlation, or estimation of statistics.

Two reports are being prepared to give partial answers to the above problems. The first report describes a noise-like signal that was analog band-pass filtered to the interval 4 Kh to 6 Kh and subsequently digitized at 20 Kh, approximately five times the rate indicated by Shannon. With these data we could use every LPth data point for LP = 2, 3, 4, or 5 and attempt to compute the value of f at the remaining data points. We then had both the measured and predicted values of f at points which were not used in the calculations. The formula used was

$$f(t) = \sum_{n = L1}^{L2} f(LP \cdot \Delta t) \cos\pi(2m - 1)t/2r \frac{\sin(\pi t/2r)}{t/2r} \quad ,$$

where

$$\Delta t = \frac{1}{20000} \text{ secs} \quad .$$

L1 and L2 were chosen so that t was near the center of the interval $(L1 \cdot LP \cdot \Delta t, L2 \cdot LP \cdot \Delta t)$. The difference L2 - L1 is the number of terms used to reproduce f at a given t and relates to the computer time required for the calculations. The sin x/x factor assures us that values of $f(n \cdot LP \cdot \Delta t)$ for $nLP\Delta t$ near t will be the heaviest contributors to the sum.

This study was initiated to provide us with some rules-of-thumb for the necessary sampling rate to yield a certain accuracy of a wave form reproduction as a function of center frequency, bandwidth and slope of the filter skirt. It was quickly discovered that the analog filter had not been properly adjusted and as a result the prediction error was large for all L1, L2, and LP. When the data were digitally filtered to the proper frequency interval, we found excellent agreement between measured and computed values for L2 - L1 as small as 6 and LP as large as 5. Some of these results are shown in Figs. 5 and 6 and the further details of this work will be reported shortly.

The second report that is in preparation is concerned with the question of further processing of signals digitized at low (near Shannon) rates. An example of the type of results presented in this report is that if a correlation function is computed from data taken at low rates and that same correlation function computed with data taken at a higher rate, then the two correlations are related via a digital filter whose response depends on the sample rate, ω_o, and B but not on the signal itself. This result permits the calculation of correlations with low numbers of data points (which is efficient) and yields, after the appropriate analog or digital filtering, precisely the same result as the calculation with more data points. No experimental data are available in a suitable format at the present time for inclusion in this Progress Report.

B. PERT

The initial intent of this project was to provide a computer system and some experienced personnel to perform PERT-TIME and PERT-COST analyses of established or proposed NAVSHIPS programs. It has become evident that a good deal more than this is desired by the contracting officer. In particular, help is wanted in establishing the programs and evaluating proposals. A study of some of the management oriented computer systems such as CPM (Critical Path Method) and linear programming was required. This study was initiated during the first quarter and has been essentially completed in that our present

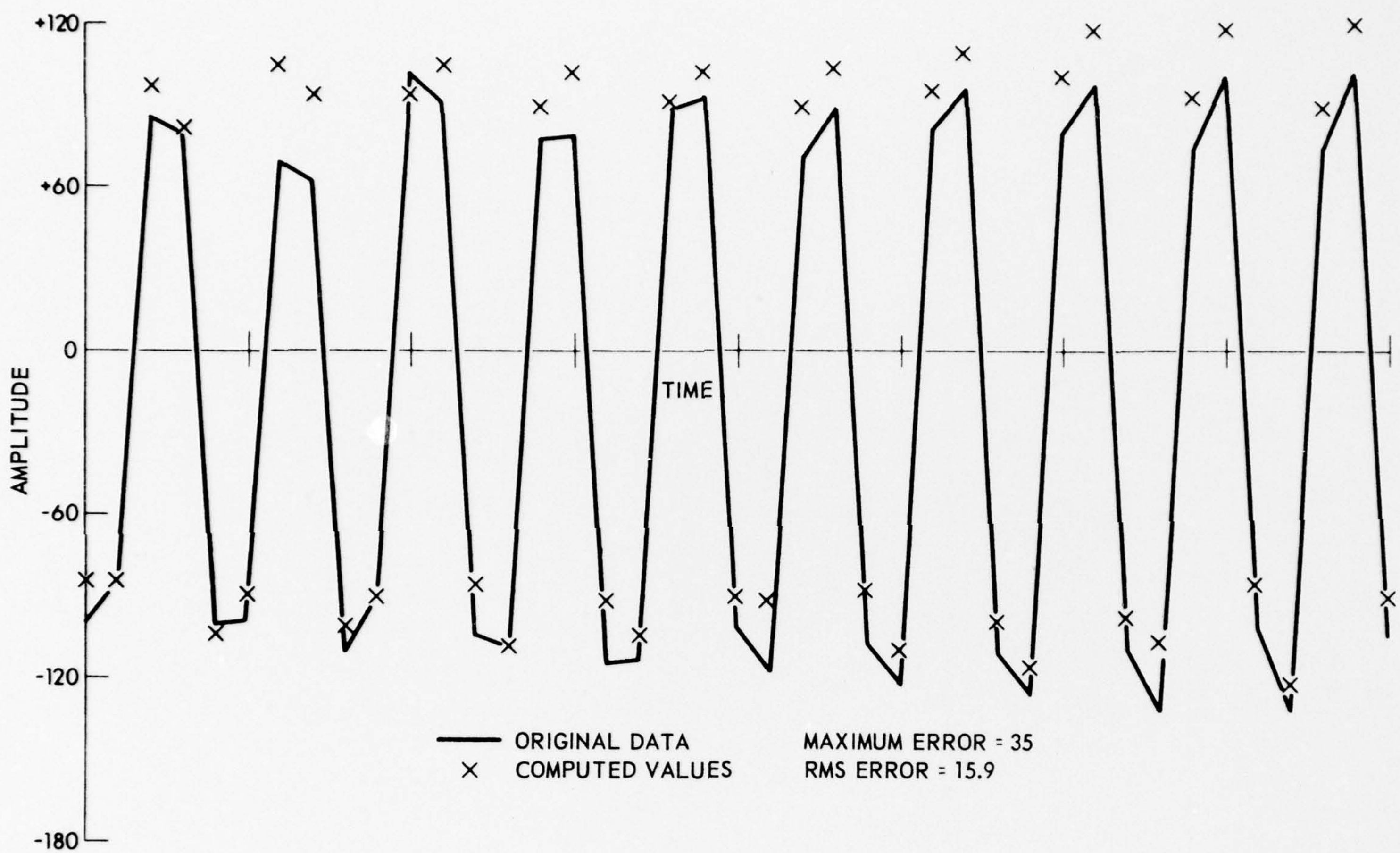

FIGURE 5
WAVEFORM REPRODUCTION
USING SHANNON'S THEOREM - ANALOG FILTER 4 kHz – 6 kHz
L2 – L1 = 30

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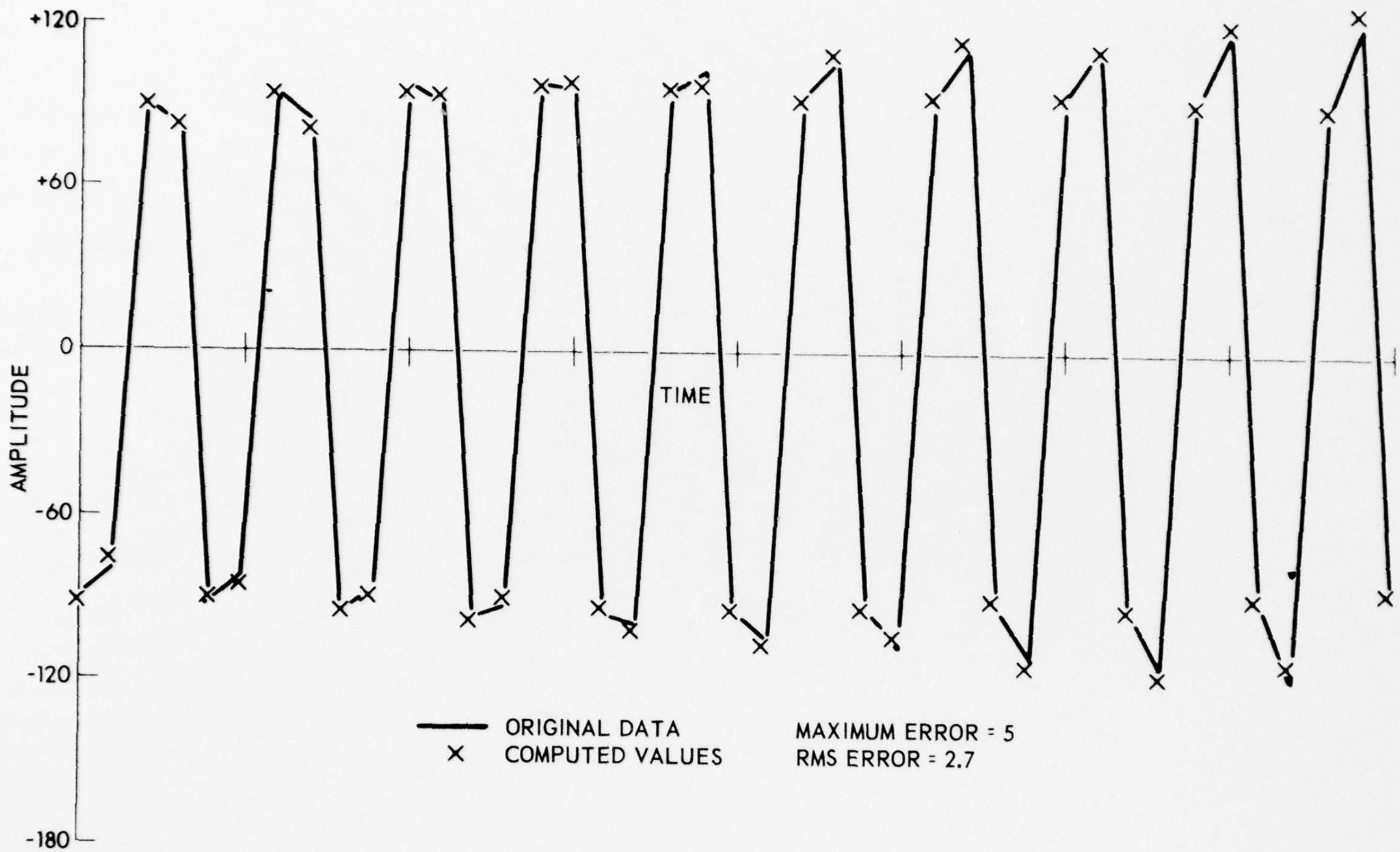

FIGURE 6
WAVEFORM REPRODUCTION
USING SHANNON'S THEOREM - DIGITAL FILTER 4 kHz – 6 kHz
L2 – L1 = 30

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estimate of the most promising approach is that of a search of all possible combinations of the individual elements.

Specifically, the exploratory development program in sonar under Code 1622 of NAVSHIPS is to be guided in the near future by eleven panels each concerned with a specific aspect of the sonar system (see Fig. 7). The various projects, both existing and proposed, under each of these panels would be evaluated on the basis of a number of, as yet undetermined, parameters such as Cost Effectiveness, Maintainability, Reliability, Number of Operators Required, Training Level, etc. The members of the panels, being among the best informed persons in the respective fields, will assign to some of these parameters a numeric value reflecting their evaluation of its merit. Some of these parameters may be computed from the panel data.

Once the data from each panel has been collected for each of the projects and proposals under its cognizance, the computer program will configure each possible final sonar system using a component from each panel. Each configuration will be assigned a numeric value (hopefully) reflecting its relative merit.

It should be noticed first that many of the items considered by each panel may not be independent as regards the final virtue of a specific unit. For example, Maintainability and Reliability should not be independently considered because a highly reliable unit that is difficult to maintain may be as useable as a less reliable unit that is more easily maintained. Secondly, there may be dependencies across panel lines. For example, it may occur that the output of a unit under one panel may not be suitable as an input to a unit of another panel.

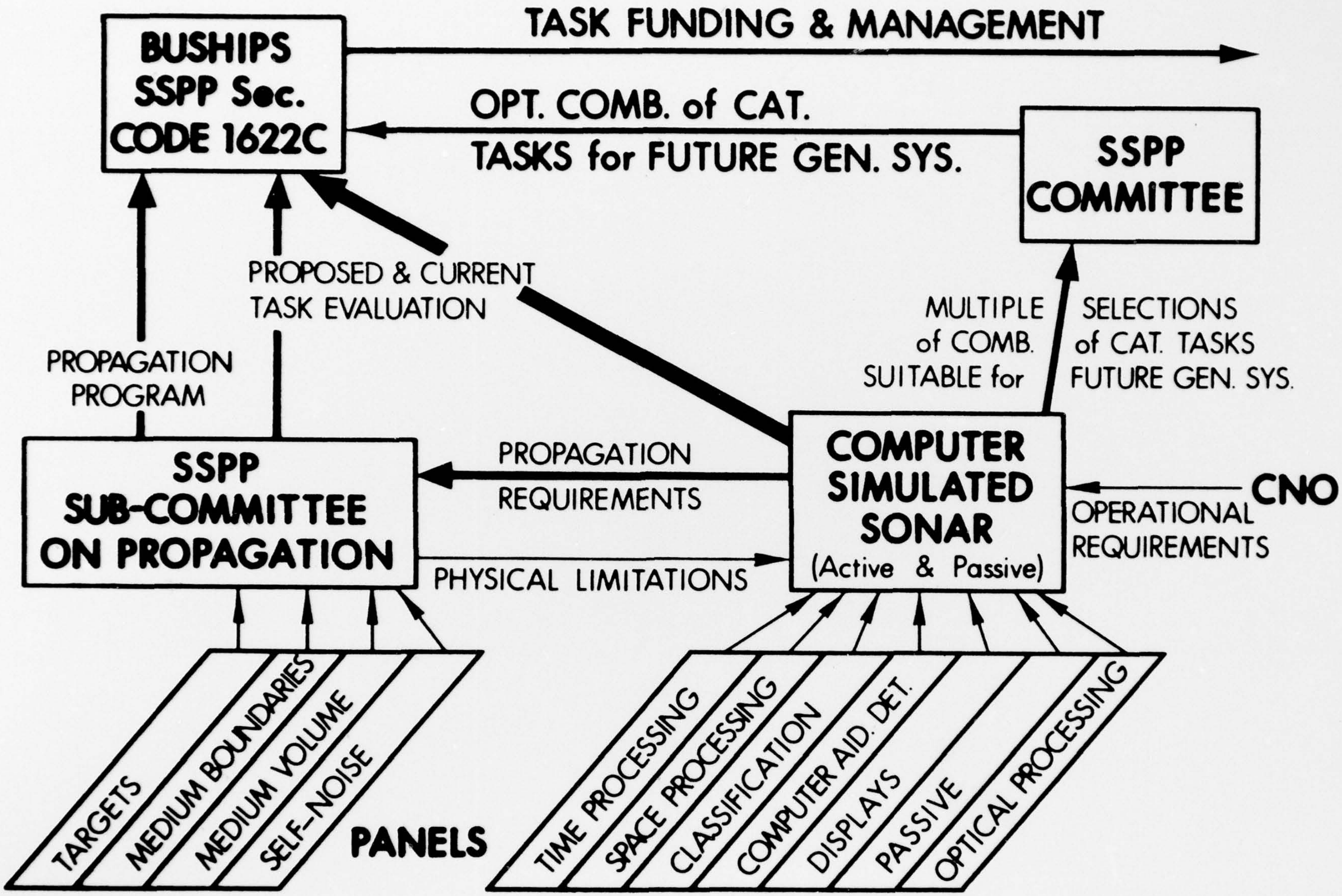

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FIGURE 7
SONAR SIGNAL PHYSICS PROGRAM - FY68

The end product might well be (1) a list of the most promising systems (for example, the best five) and (2) an indication as to which units considered by the individual panels show the most promise. The output could be prepared in the forms shown in Figs. 8 and 9.

At present it is not intended that any decision mechanism be included in the program. The indicated results would be made available to the panels and to NAVSHIPS personnel as an aid to the human decision process. This "human touch" is recommended at least until the computer, the program and panels are proved infallible.

A great deal of the programming is not required to implement this evaluation system. It cannot, however, be carried much further without some decisions being made with regard to the parameters of interest, data format, relative importance of parameters, etc. In the final analysis, these decisions must be made by the panel members and/or NAVSHIPS personnel. One reason for our continued though limited interest in SONARCON is that it may serve as an excellent base on which to build the required programs. The SONARCON section of this report should be referenced here to expand on the above statement.

The success or failure depends on (1) establishing a suitable format for communication between the panels and computer and (2) the degree to which the individual panels can evaluate the various projects under their tutelage. One of the difficult problems the NAVSHIPS personnel will have to deal with will be evaluating the output of this program in light of the above mentioned limitations.

C. SONARCON

On 12 January 1967 a meeting was held at the offices of TRACOR, Incorporated in Austin to discuss the transfer of some programs written by the TRACOR staff for the CDC 3200 computer to DRL. The attendees at this

RATING 0 TO 1	PANEL 1	PANEL 2	PANEL 3	.....	PANEL 11
0.63	UNIT 3	UNIT 1	UNIT 4	.....	UNIT 1
0.60	UNIT 3	UNIT 2	UNIT 1	.....	UNIT 3
0.59	UNIT 2	UNIT 1	UNIT 3	.....	UNIT 3
. . .					
0.50	UNIT 1	UNIT 2	UNIT 1	.....	UNIT 2

NEXT RATING 0.45
AVERAGE RATING 0.30
TOTAL NUMBER OF SYSTEMS RATED 5000
TOTAL NUMBER OF SYSTEMS WITH ZERO* RATING 2000

* ZERO RATING IMPLIES INCOMPATIBILITY BETWEEN UNITS

DRL - UT
DWG AS-67-457
GSI - RFO
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FIGURE 8
SAMPLE SYSTEM REPORT CODE 1622

	HIGHEST RATING	LOWEST RATING	AVERAGE RATING	NUMBER OF ZERO RATED SYSTEMS
UNIT 1	0.50	0	0.41	500
UNIT 2	0.59	0.09	0.43	0
UNIT 3	0.63	0	0.30	1000
UNIT 4	0.41	0	0.25	980
UNIT 5	0.45	0.20	0.31	0

TOTAL NUMBER OF SYSTEMS RATED 5000

TOTAL NUMBER OF SYSTEMS WITH ZERO RATING 2000

FIGURE 9
SAMPLE REPORT TO PANEL 1 CODE 1622

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DWG AS-67-458
GSI - RFO
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meeting were Messrs. Bob Courts and J. C. Humphrey of TRACOR and G. E. Ellis and G. S. Innis of DRL.

Mr. Courts expressed the opinion that the only programs for the 3200 that were sufficiently well documented to be usable were those in the SONARCON system. He also suggested that he would "install" that system at DRL free of charge if we would let them use our computer to prepare for an installation at NADC. This was agreed upon and the installation took place during the weeks of 16 and 23 January 1967.

Table I contains a fairly complete list of the user routines delivered with the SONARCON system. The utility of this system is presently very much in doubt because of a shortage of information on both the routines and the system.

The write-up of some of the routines refer to TRACOR publications which are not available, and some of the routines apparently have errors in them. It is also true that SONARCON is very slow running.

In order to speed up the operation under this system, an attempt has been made to install SONARCON on the DISK at DRL. This has not been successful yet, but probably could be achieved if sufficient effort were applied. This brings us to the crux of the whole problem. SONARCON is a simulation language which may or may not be well suited to our general needs. The problems of becoming familiar with, maintaining and improving (updating) such a system are large, particularly in a "home brew" like SONARCON. This package evolved as TRACOR's needs arose and probably served their purposes very well. DRL has arrived on the scene under an entirely different set of conditions and, in particular, there are a large number of simulation languages available in the computer industry. Some of these have received the attention of a staff of experienced computer scientists. As a result they are more general, more easily maintained, and better documented than SONARCON. They may not be as well suited to NAVSHIPS problems as the

TABLE I

SONARCON Subroutines-
Black Boxes
CDC-3200

NAME	DESCRIPTION OF PROBLEM
ACLIP-	takes an input time function and clips all samples which are in absolute value less than an input threshold value.
ACOR-	given a lag offset L and an input function X_i of length N, will form the L-1 values of the autocorrelation function $X_i \cdot X_i$.
ADPAV-	modifies a time function replacing samples by new values, middle value given by field from ENGSP.
ADRG-	find a running average of L words of an input time function.
BETA-	performs bandwidth measurement using the input and output data from the topology chain LCORR, RECT, and AVRG--pass input and output through SGMN.
BROWN-	a modification of SRTPC--in the horizontal axis scale and computation of x_i points.
CHTAP-	allows user to request a tape change within his topology without having to call SONARCON.
CLEAR-	sets a field of N words to zeros.
CLECT-	collects individual output fields of LOCAL into a single field.
CLIP-	takes input time function, sets all positive samples to +1 and all negative samples to -1.
CNTPL-	counts the number of digital pulses occuring in one input file of time function data.
CODE-	produces output field containing N frequencies $F_o + i\Delta F(i=0, 1,...N-1)$ arranged in random order-given initial frequency F_o, change of frequency ΔF, and integer N.
COOK-	correlation on-time function data are obtained through use of COOK, AVG, RECT, and ORL.
CORGN-	generates a correlator reference in either clipped or unclipped mode.

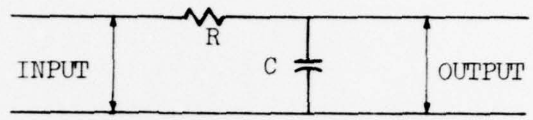
NAME	DESCRIPTION OF PROBLEM
CORR-	used to perform linear dc replica correlation with a clipped reference.
CUMPR-	uses output field from ACOR to compute a cumulative power spectrum and optionally prints and/or graphs results.
DBPRC-	dB processor-perfoms dB analysis on-time functions.
DEMPX-	given input time function, representing multiplexed data, demultiplexes the data producing a specified number--nine or fewer--output time-functions.
DLETE-	given input time function X_i and threshold T, deletes from the input function every sample X_i that is greater than or equal to T.
DOP-	given input field and input value V, adds V to every element of the input field.
DPRNT-	given input field length N, contains either single or double precision binary scaled numbers, produces output field of floating point numbers--optional print out.
DSPLY-	processes the output field of CLECT to obtain a condensed printer display of the results.
ECOEX-	given a reference point and time function, extracts a given number of samples, usually in the vicinity of an echo--outputs as a file in a time function.
ENGSP-	uses two input time functions X and Y to compute one output time function Z where $Z_i = \sqrt{X_i^2 + Y_i^2}$
EXNSE-	modification of ECOEX, extracts given number of noise samples usually before and after the echo (omitting the echo)--output as a file in a time function.
FAC-	a false alarm counter--uses output time functions from LPEAK as input--uses same algorithm as SORT.
FCORR-	computes and prints correlation between two fields--both output from FORTM and same length.
FILBK-	takes one file of input time functions and breaks it into N files of output time functions.

NAME	DESCRIPTION OF PROBLEM
FLDAL-	takes field of indices from RIGEN and shifts locations of peak values so that peaks are in same location in each field.
FLDAV-	computes the mean and standard deviation of corresponding samples in K input fields each of length N.
FLDX-	accepts input field length N and breaks it into output fields of length K, the user's input value.
FORDV-	has four channels of input fields and produces two channels of field output.
FORIN-	given two fields of input, representing a real and an imaginary part of a Fourier transform, computes the time function corresponding to the inverse transform of the input fields.
FORK-	causes a partial record fork if the first sample of its input record of time function is zero, else proceeds normally.
FORKA-	given input integer N, causes a partial record fork for N records, after which it allows control to proceed.
FORMU-	modifies four input fields into two input fields.
FORTM-	generates a field of power constants by analyzing time function data--used for power spectra or reverberation analysis.
FORYA-	computes the Fourier transform of a time function and generates two fields of Fourier coefficients.
FRQAL-	computes a frequency analysis of an input time function.
FSET-	enables user to set a field of variable length to values specified on standard SONARCON data cards.
GEN-	given output field from CODE, a sampling rate ΔT, and a number K of samples in any given frequency change, will generate the PR pulse defined by the N frequency.
HETRF-	generates two heterodyne correlator references.
IDENTIFY-	sets up for printing an identification line.
INSAM-	given L values of normalized autocorrelation function computed by ACORP, estimates the number of samples per independent sample v and the correlation time τ.
INVRT-	takes a magnitude correlator reference, inverts the order, and produces a magnitude correlator reference.

NAME	DESCRIPTION OF PROBLEM
LAG-	designed to provide a specified number (N) of empty records.
LCORR-	takes a clipped correlator reference and correlates it against an input time function.
LOCAL-	for two given integers N and K will determine the N largest samples in a file of input time function such that at K samples lie between each of the N largest samples.
LOCPR-	used to print out the results of LOCAL.
LPEAK-	selects all local peaks from an input time function and gives two time functions out.
MAPST-	a SONARCON diagnostic program; it causes the subprogram MAP.STG to be executed as called for in the topology list--prints locations of stored records.
MAP.STG-	inspects, edits, and prints out the contents of all packed and unpacked storage at any given time during the running of SONARCON.
MCORR-	takes a correlator reference and correlates it against an input time function. Output is a time function the same length as input.
MFFL-	simulates a Monostable Multivibrator.
MINUS-	has an input one time function, whose elements are X_i, and as output one time function whose elements are Y_i.
MIXER-	used to accomplish multiplication of a time function with $\sin(2\pi ft)$ or $\cos(2\pi ft)$.
MPX-	multiplexes the data from eight or fewer input time functions and outputs a single time function, which is the composite of the input data.
MRFEX-	may be used to convert sections of a time function to field format suitable for use by MCORR.
MTEOF-	causes file marks to be written on the output tape where several output files are generated for one input file.
MTI-	controls the reading of time functions from magnetic tape within SONARCON.
MTIF-	reads a field from magnetic tape within SONARCON.

NAME	DESCRIPTION OF PROBLEM
MTO-	controls the writing of time functions onto magnetic tape within SONARCON.
MTOF-	writes a field on magnetic tape within SONARCON.
MULL-	given two input time functions, will produce an output time function, each element of which is the product of the corresponding elements of the input time function.
NCORR-	correlates an input time function against a given reference and produces a normalized magnitude correlation function as output.
NORM-	takes an input time function and normalizes each point of the function relative to the mean and standard deviation of sections of data on either side of the point.
NORMA-	takes an input time function and normalizes each point of the function relative to the mean of sections of data on either side of the point.
ORDER-	given input similar to that produced by CODE, will produce an output field in the same order as the input field but relative to two new input values, F'_o & $\Delta f'$.
ORKIN-	will output M time function samples and skip N samples to the end-of-the-file.
ORL-	outputs the largest of the corresponding words of two input time functions.
PARTO-	partitions the first input time function specified and partitions all other input channels at the same point.
PEAKP-	examines center point of an interval and shifts interval of integer length 2T+1 along an input time function.
PHASM-	given set of samples representing wave periods and scales between 0 and 5 volts, converts these to frequencies scaled between +5 (max. freq.) and -5 (min. freq.).
PIP-	given two integers N and K and two sample values V_1 and V_2, generates an output time function, the first N samples of which have the value V_1.
PRNT-	will print a field of numbers in either octal or decimal format.

NAME	DESCRIPTION OF PROBLEM
PSCAN-	obtains a peak value from a section of length N of input time function and produces an output time function of 512 such peak values.
RCFIL-	may be used to simulate a low pass RC filter as shown:



NAME	DESCRIPTION OF PROBLEM
RECT-	input of one time function; finds the absolute value of each word in the time function and outputs these values.
RED-	given output of CLECT, and an integer L($\leq$ N), will consider the N - L + 1 sections of L consecutive echo cylcles and filter out those samples that do not contribute to a track.
REDUC-	adds a reference field to a segment of each file of an input time function.
REJ-	gives an output every Nth sample of the input time function where N is an input value.
REP-	generates the expected rectified echo images with aspect as a parameter using those structural features most likely to appear.
REVRB-	computes a reverberation analysis on an input time function.
RFEXT-	takes a time function input and provides a correlator reference field as output.
RIGEN-	given successive input fields, prepares one output field.
RUNS-	computes the means and standard deviations of contiguous sections of an input time function.
SCAT-	is used to convert a correlation reference into a string of instructions suitable for use by CORR to accomplish linear dc replica correlation with a clipped reference.
SCEXT-	is used to extract a section of data from an input time function. The last record is zero filled.
SCHL-	simulates a Schmidt trigger.
SCLE-	scale input time functions by a scale factor specified on data cards.

NAME	DESCRIPTION OF PROBLEM
SCSCN-	scans specified equally spaced sections of an input time function for the peak in each section.
SGHID-	hides an input signal specified number of times in an input noise file.
SGMN-	used to determine the mean and standard deviation of a specified number, N, of words of input time function.
SGMNF-	computes and prints the mean and standard deviation of an input field.
SINER-	generates a continuous sine wave with a mean of zero, standard deviation of .707.
SIT-	sets a field of specified length to a specified value.
SMRAT-	computes the ratio of the standard deviation to the mean from two equi-length input fields.
SORT-	takes an input time function and sorts this data into a specified number of counting locations that are provided by the SET black box.
SRTPC-	computes a statistical analysis and if desired gives a graphed output.
SSGEN-	is a sine signal generator and outputs the signal as a time function.
STIME-	sets a threshold value and measures the time between successive increases in time function values over this threshold.
TFP-	is used to plot time function output and simulates Sanborn output.
TINTG-	sums the points of an input time function and outputs a time function representing an incremental integration.
TMUL-	multiplies corresponding elements of two input time functions, where the second function can be delayed a specified number of samples, and outputs these products as a time function.
TRUST-	outputs every Nth word of an input field forming a new field of length K and this process must be repeated N times for all words of the original field to have been used.

NAME	DESCRIPTION OF PROBLEM
TSUM-	two input time functions are added together with the capability of delaying the second function by K samples.
TWOSQ-	takes two fields of input having formats the same as the output from FORYA or FORDV and produces a field of power constants suitable for use by PWRSB.

SONARCON system appears to be. *The point is that DRL must choose one of* these simulation languages in which to develop her expertise and ignore all others. To make this choice, a more detailed study of some of the other simulation languages should be conducted. We do not feel that an extensive effort with SONARCON is warrented without this study.

During the next quarter we will pursue this study as well as continuing our introduction to SONARCON in the hope of being able to make a more definitive statement about the relative merits of these languages.

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